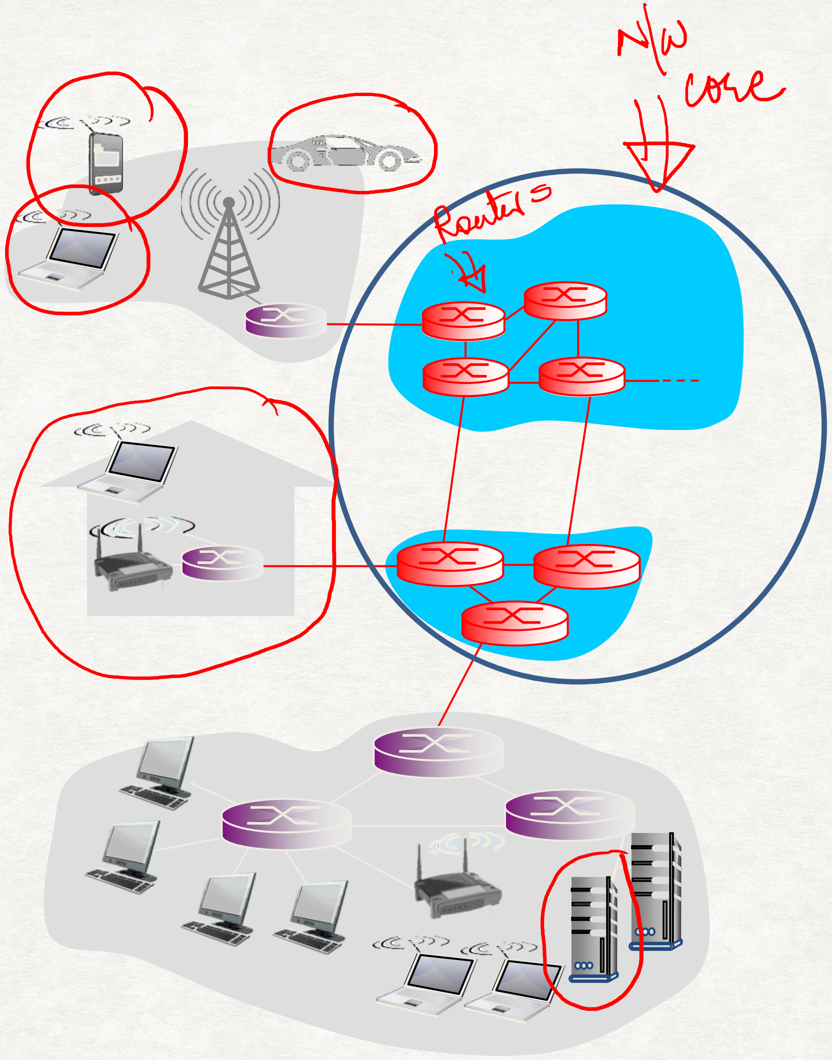
Networking II: Basic Architectural Ideas & Principles

COMPSCI110

The Network Core

- Mesh of interconnected routers (switching devices).

- How is data transferred through a network?
  - Circuit-switching Technology → PSTN (Telephone)
  - Packet-switching Technology → Internet

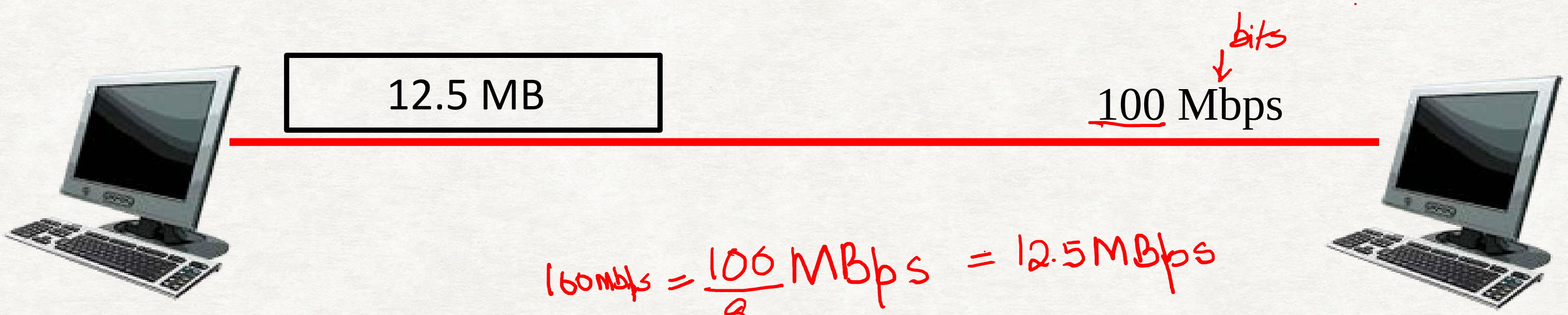


Reference : *Computer Networking:A Top Down Approach by Kurose & Ross*

Sharing a communication channel

Say we're wanting to send a large file between two hosts

- E.g., file is 12,500,000 bytes (12.5 MB) in size
- This file size is typical for, e.g., a high resolution digital camera photo



$100\text{Mbps} = \frac{100}{8}\text{MBps} = 12.5\text{MBps}$

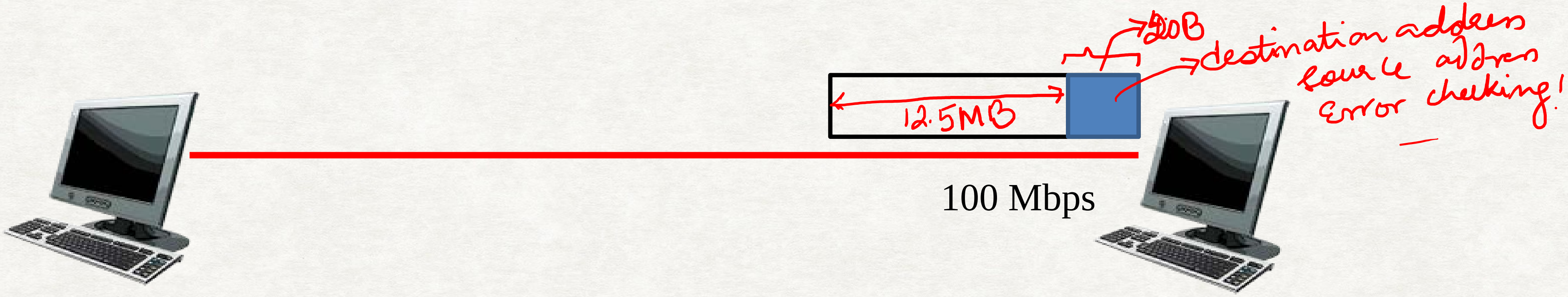
100 Mbps = 12.5 MBps ("megabytes per second")

$\frac{12.5\text{MB}}{12.5\text{MBps}} = 1$ second

Communication attempt 1

Method: Send the 12.5 MB file in one piece with a continuous transmission.

Need to send a little information at the start as to where the file needs to go etc., but this doesn't take long (needs just a few dozen bytes: negligible time).

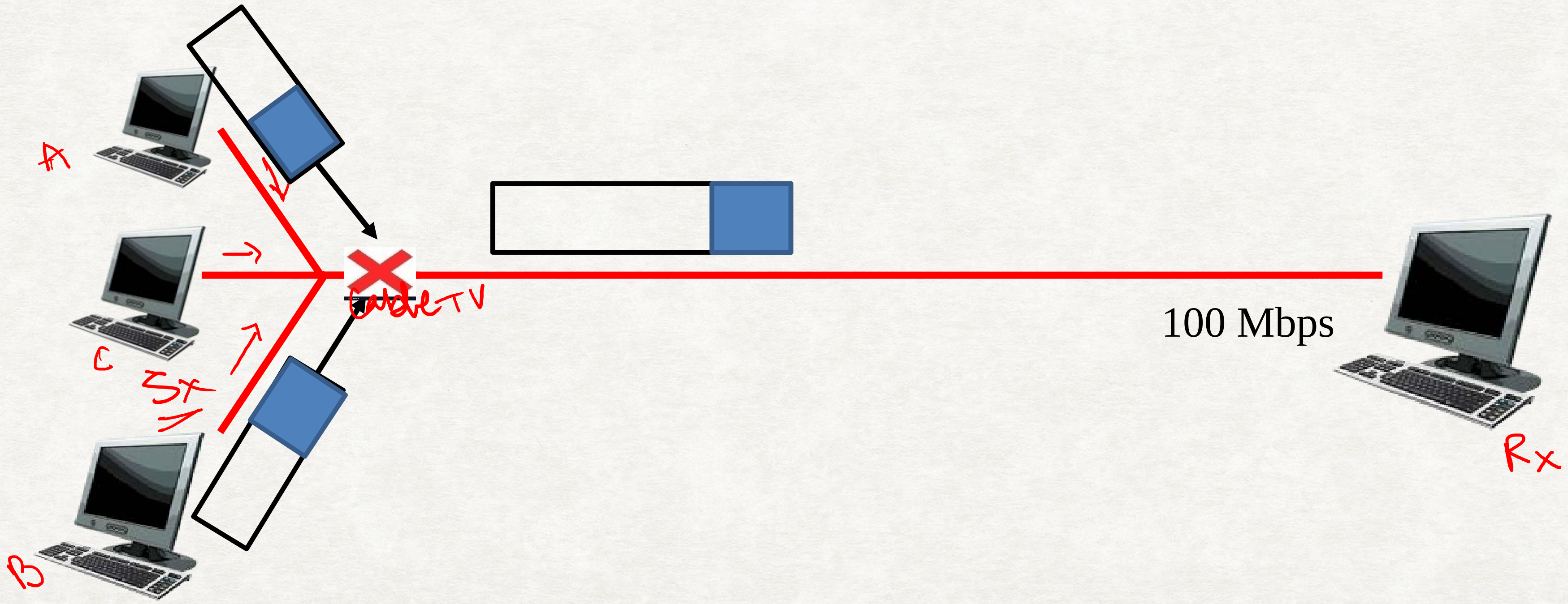


Sending the file itself takes 1 second.

Communication attempt 1

Result: ~~12.5 MB~~ file transferred in 1 second

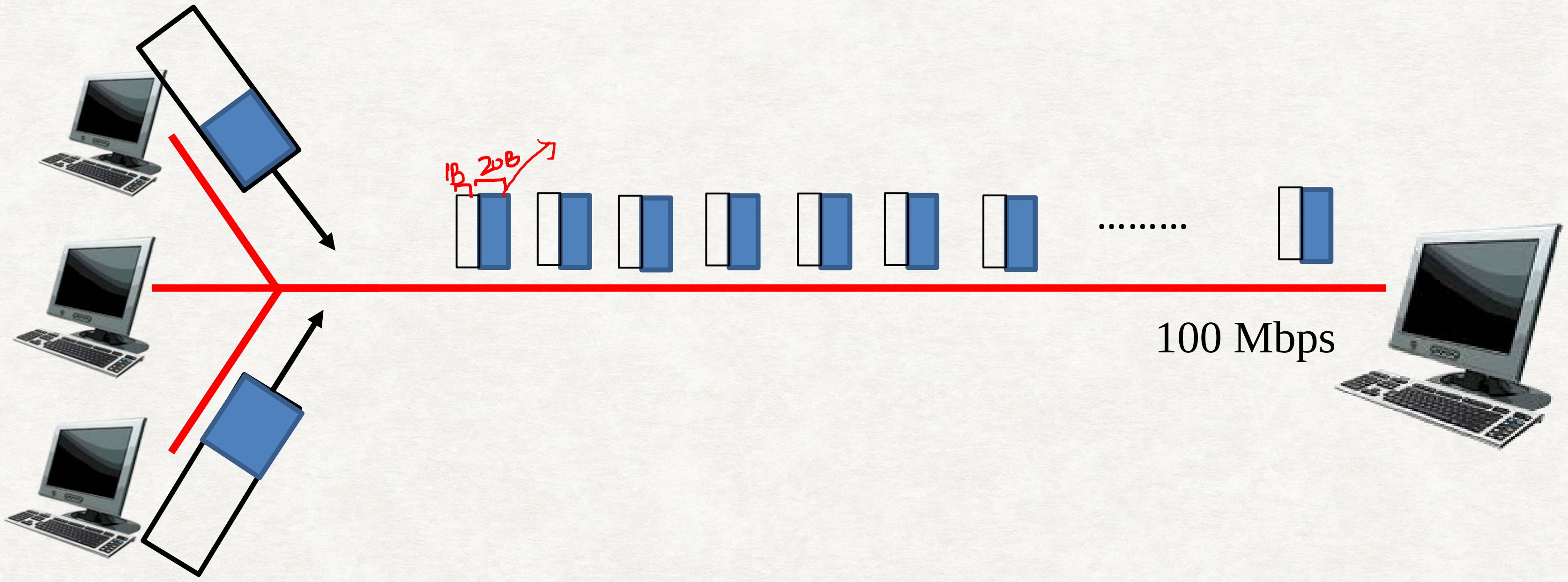
Is there a problem here?



Communication attempt 2

12.5MB

Method: Take the other extreme and send the file byte by byte with breaks in between.



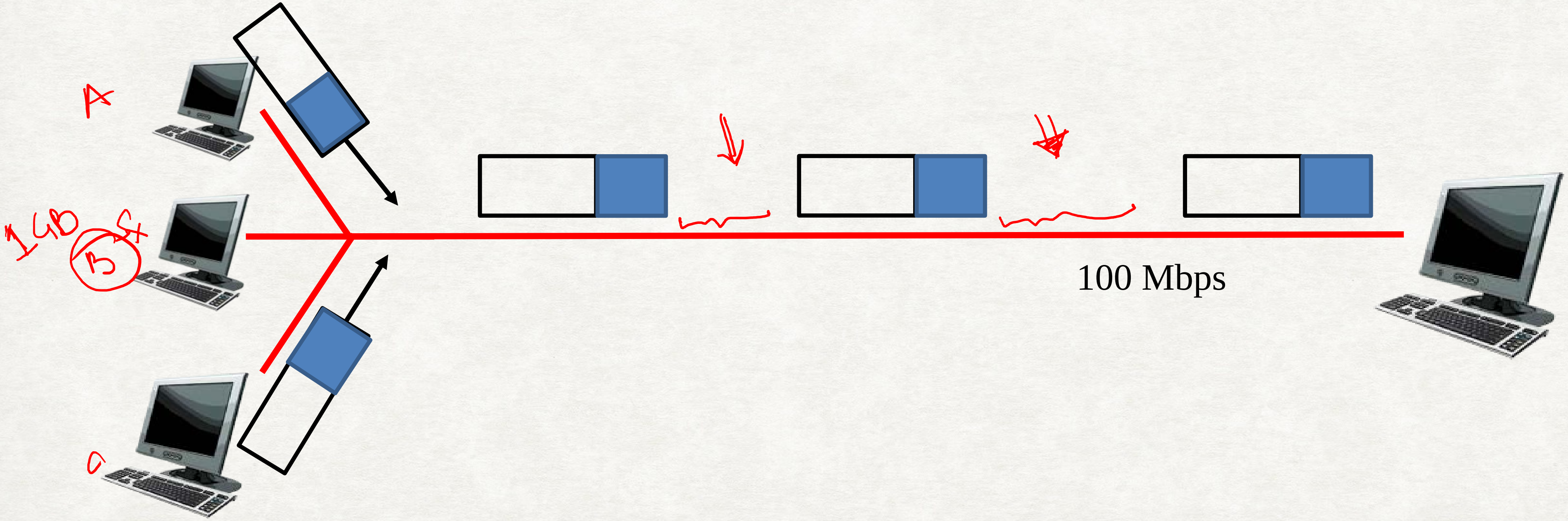
Smarter communication attempt

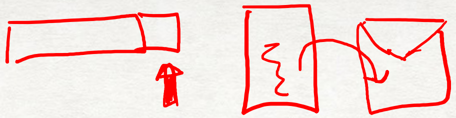
$$\frac{12.5 \times 10^6 \text{ bytes}}{1250} = \underline{\underline{10,000 \text{ chunks}}}$$

Method:

1500 bytes
1500 bytes

Split the file up into chunks of 1250 bytes. That makes 10,000 chunks.
Leave breaks between the chunks to allow other hosts to jump in if needed.

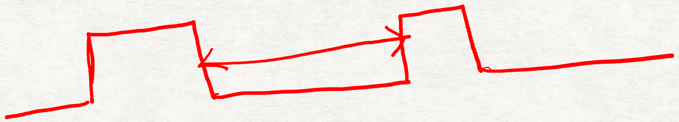




Packet Switching – one of the core principles of the Internet

- This idea of breaking up our data into discrete **self-contained** chunks is called packet switching.
  - A packet is self-contained since it carries information necessary for it to reach its destination.
  - Packets *are forwarded* from one router to the next, across links on path from source to destination.
  - Each packet is transmitted at full link capacity.

- Data traffic is bursty



  - Packet Switching allows flows (different packets from an application) to share link capacity effectively.

Alternative technique: circuit switching

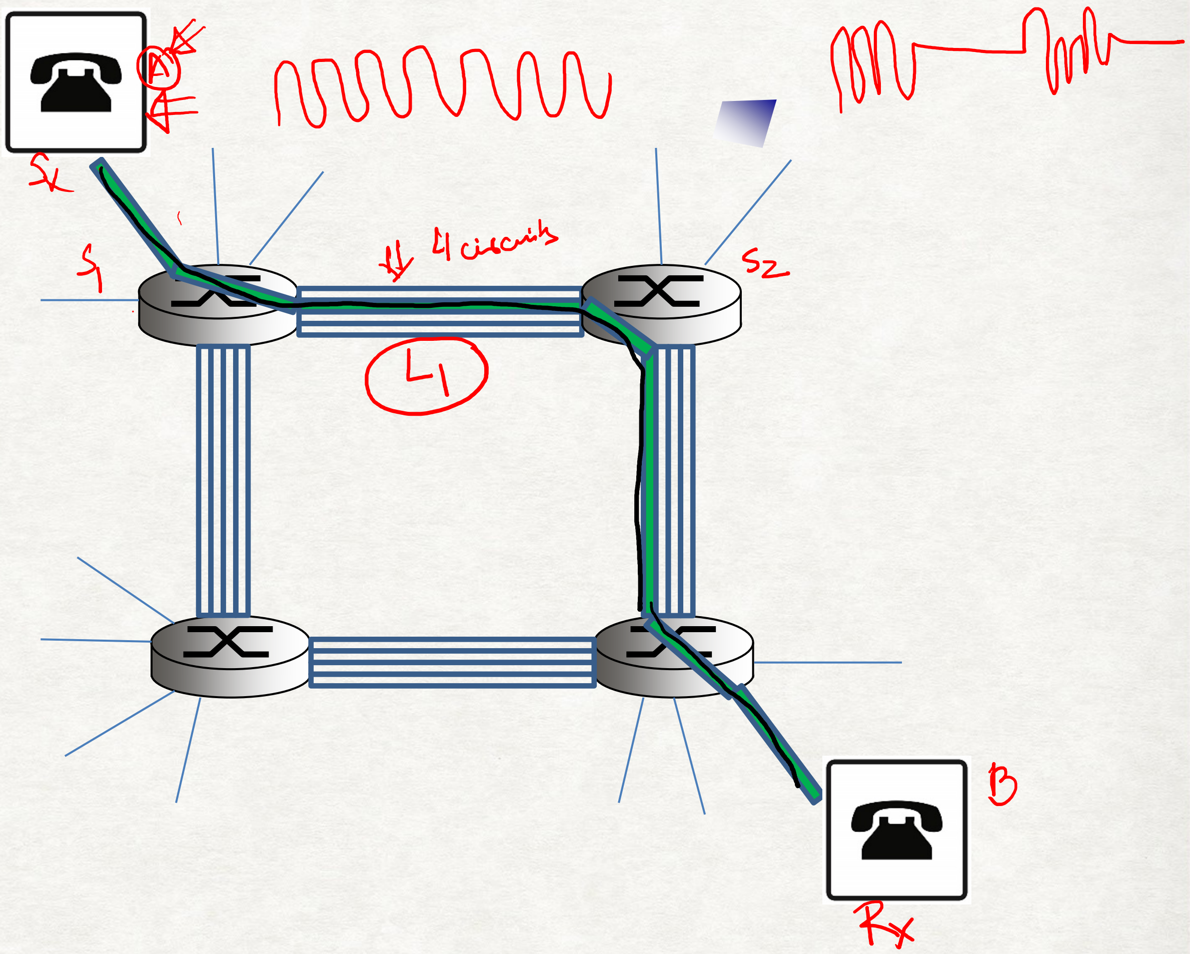
End-end resources allocated to, and reserved for "call" between source & destination.

Dedicated resources: no sharing

- circuit-like (guaranteed) performance

Circuit segment idle if not used by call *(no sharing)*

Commonly used in traditional telephone networks



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Circuit switching: FDM versus TDM

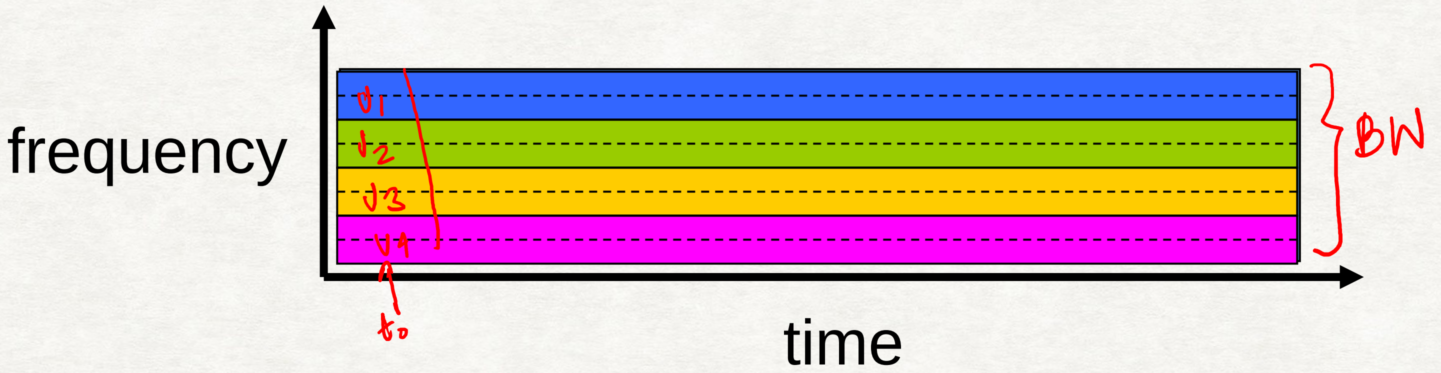
A circuit in a link is implemented with

- Frequency Division Multiplexing (FDM)
- Time Division Multiplexing (TDM)

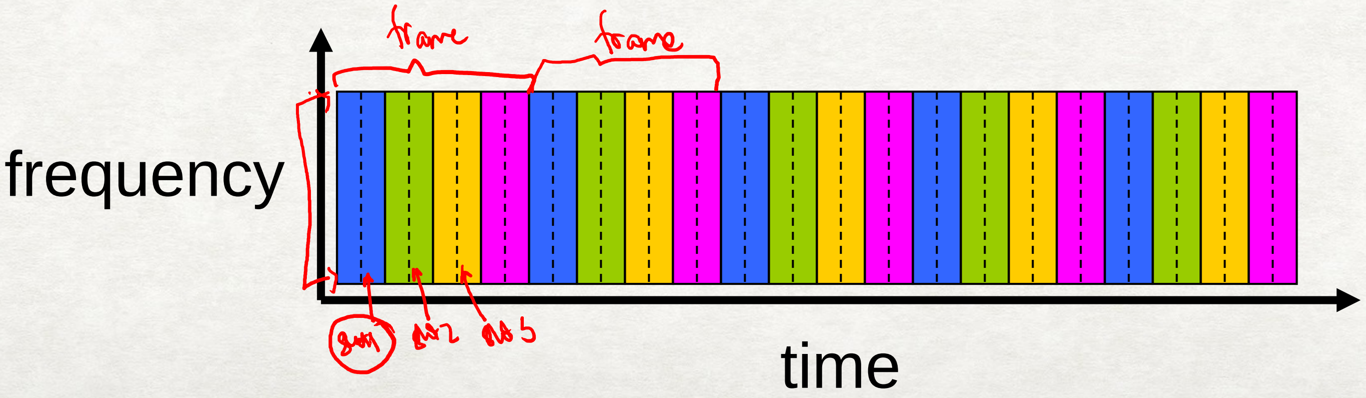
Example:

4 users

FDM

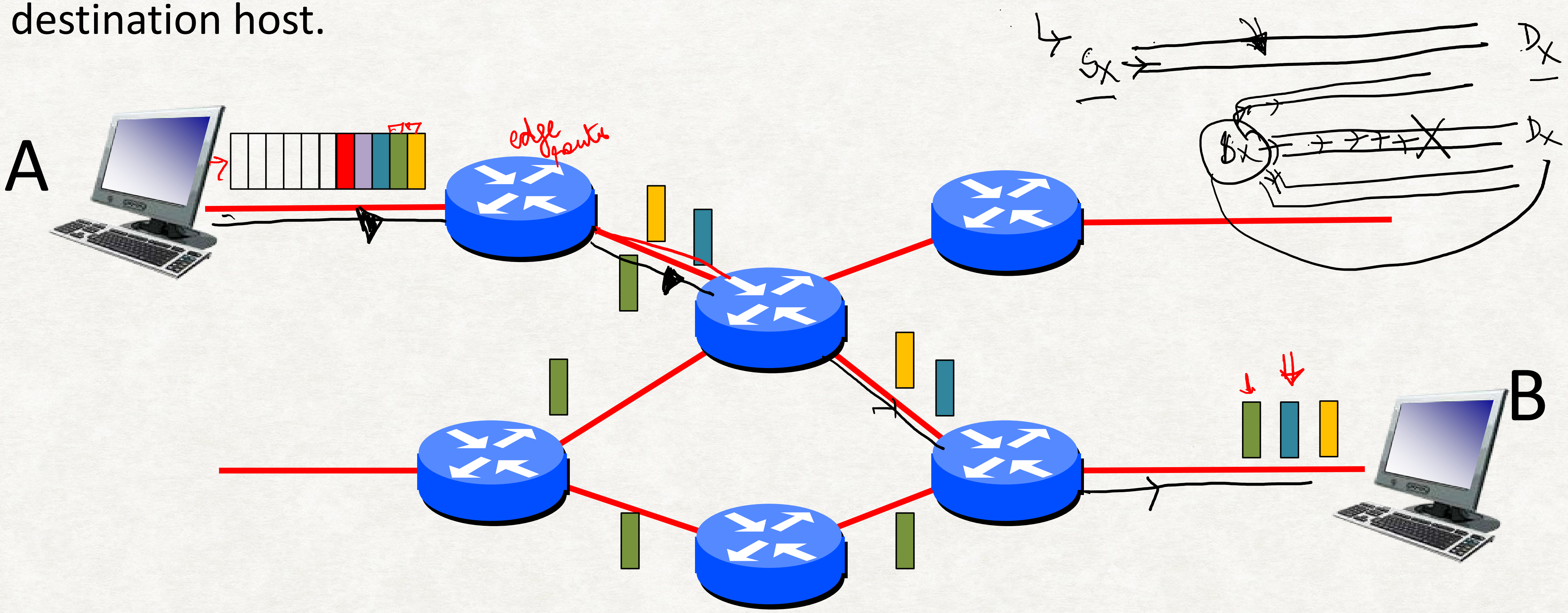


TDM

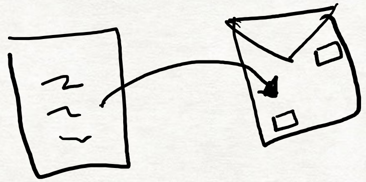


Packet Switching Technology

Each packet in the message travels independent of the other packets to the destination host.



Network Packets



Packets consist of TWO parts

destination
→ source

Packet header (address & metadata)	Packet payload (data we want to transfer)

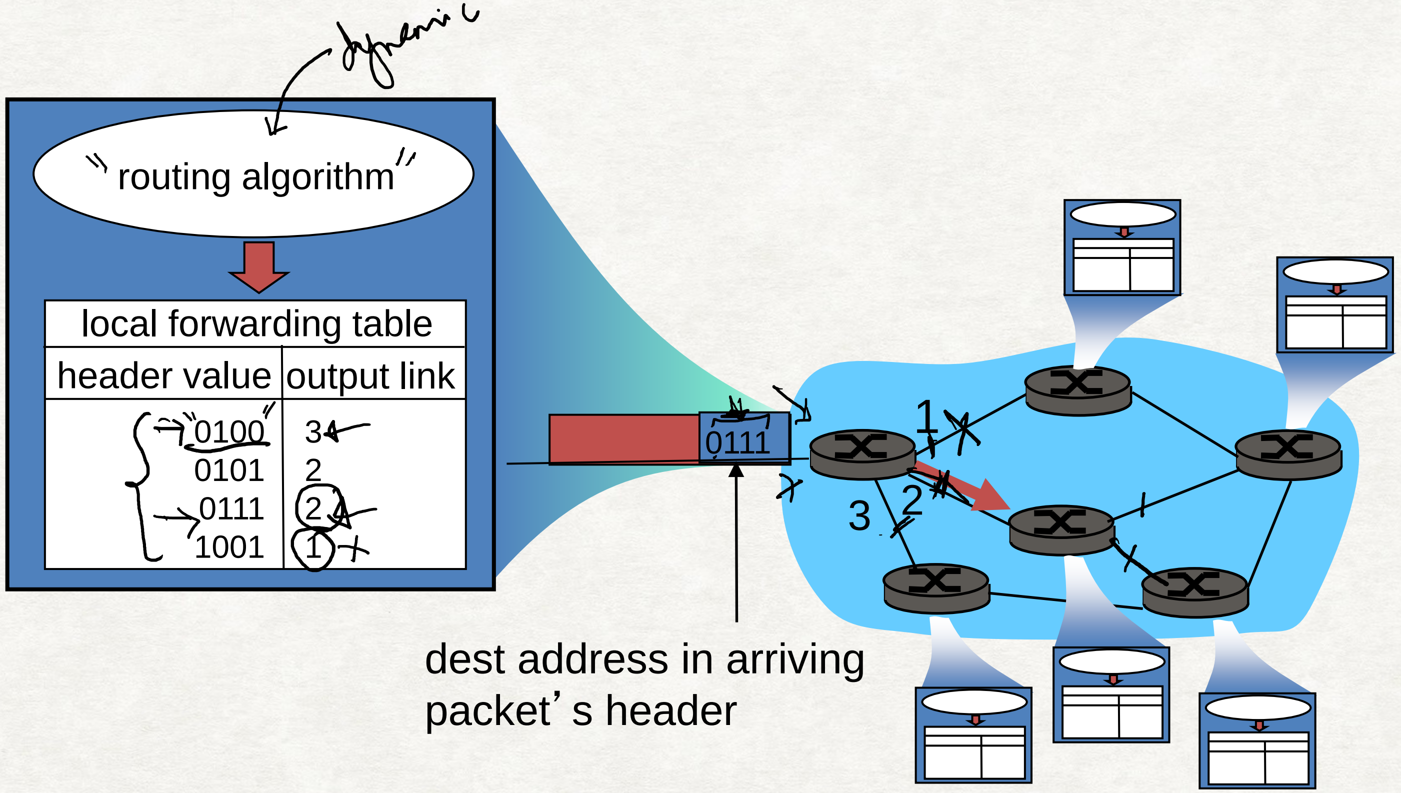
We say that the "header" ***encapsulates*** the payload.

In some cases (e.g., Ethernet frames), there may also be a trailer after the payload.

Packet header (address & metadata)	Packet payload (data we want to transfer)	Packet Trailer

The total size of the packet and the size of the payload are limited.
E.g., on Ethernet the Maximum Transmission Unit (MTU) is 1500 bytes.

Switches/Routers (in the network's core) use the address in the packet to determine how to forward the packet based on a routing table.



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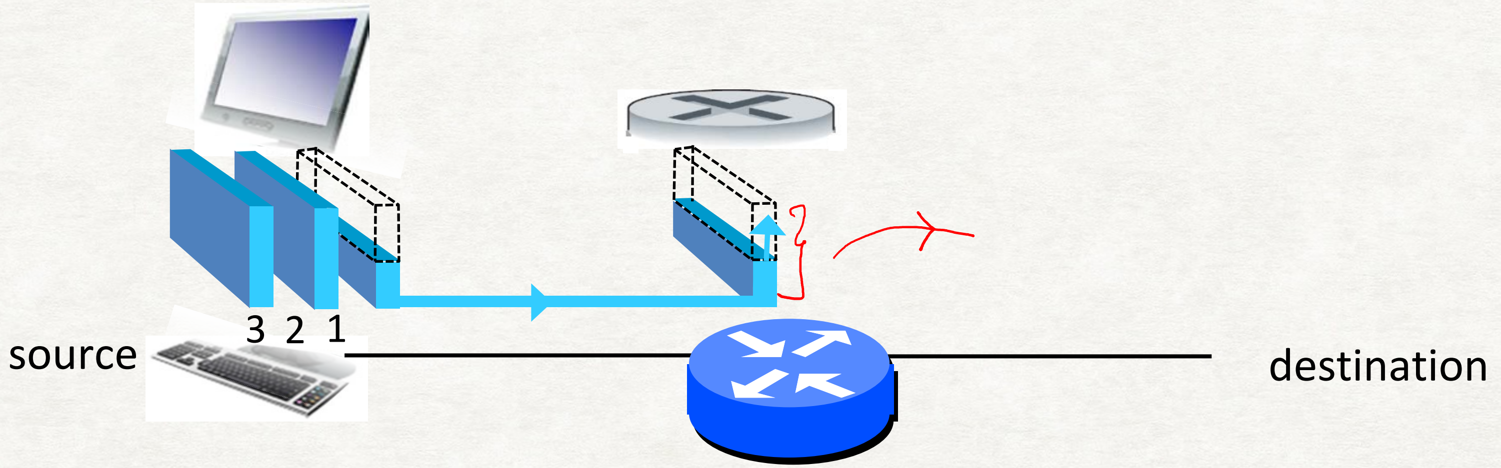
Packet-switching: store-and-forward



Store and forward: entire packet must arrive at router before it can be transmitted on next link

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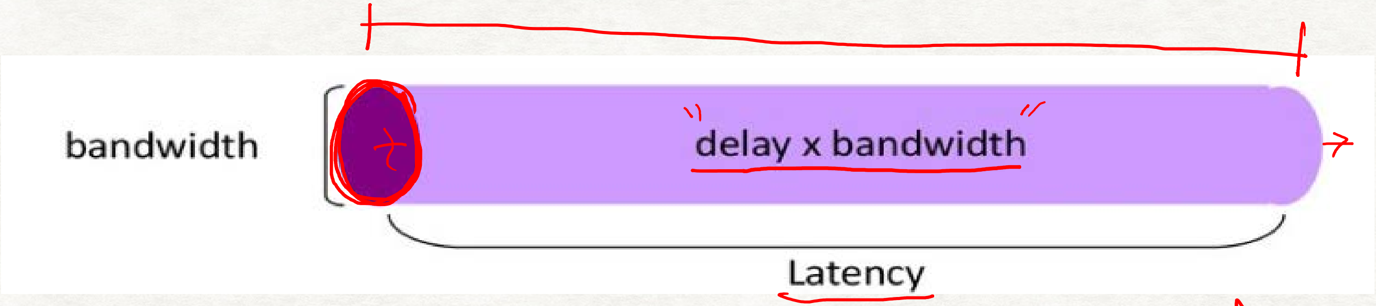
How long does it take a packet to travel on a link?

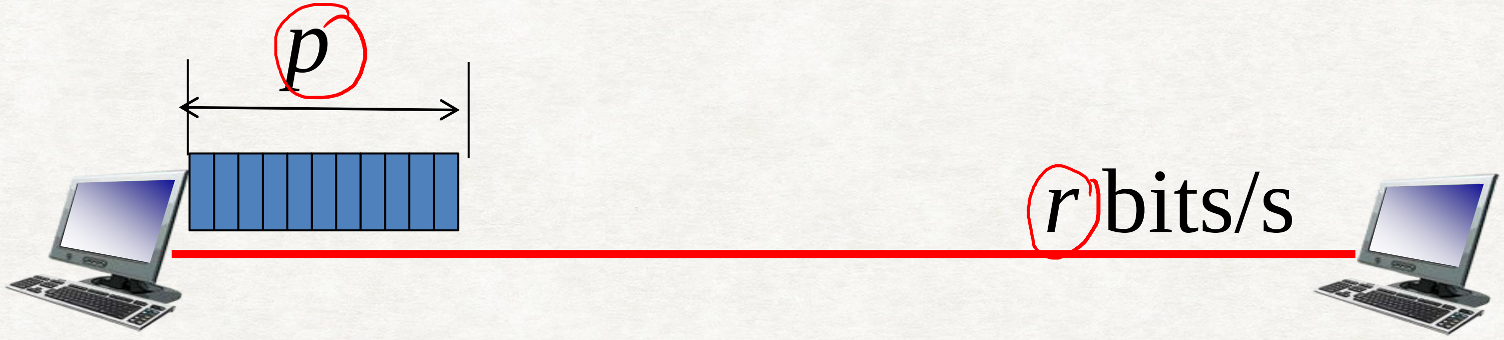


Transmission speed

1. Capacity of the link (or, Bandwidth)
   - bits/sec or bps

2. Length of the link (Latency):
   - propagation time of a bit

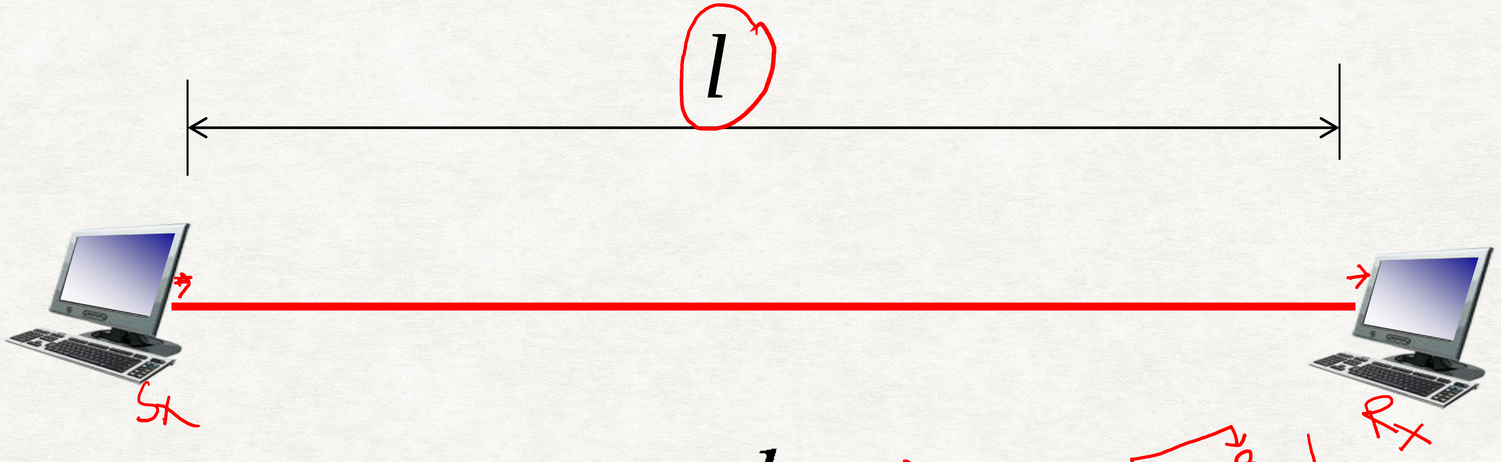
Bandwidth of a link determines the transmission delay of the packet.(or sometimes called the packetization delay)



$$t_p = \frac{p \text{ (bits)}}{r \text{ (bits/sec)}}$$

Packetization

Length of a link determines the propagation delay (time it takes for a single bit to travel over the link).

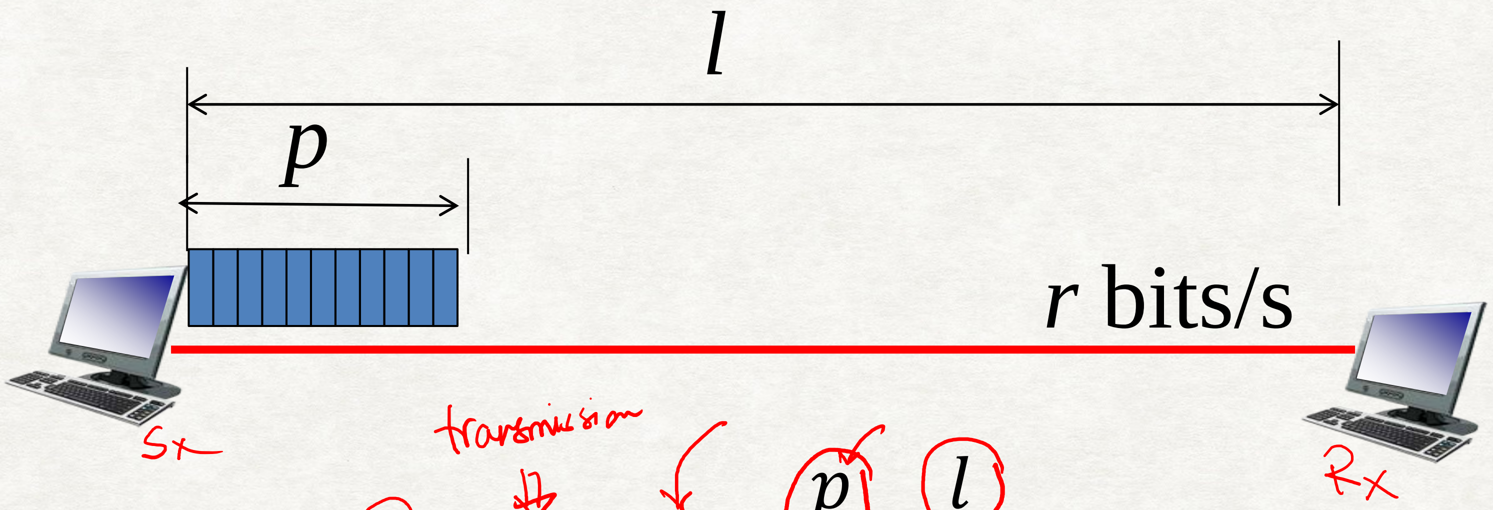


$$t_l = \frac{l}{c}$$

$\boxed{2\times10^8\ \mathrm{m/s}} \rightarrow 3\times10^8\ \mathrm{m/s}$

Total time to send a packet across a link

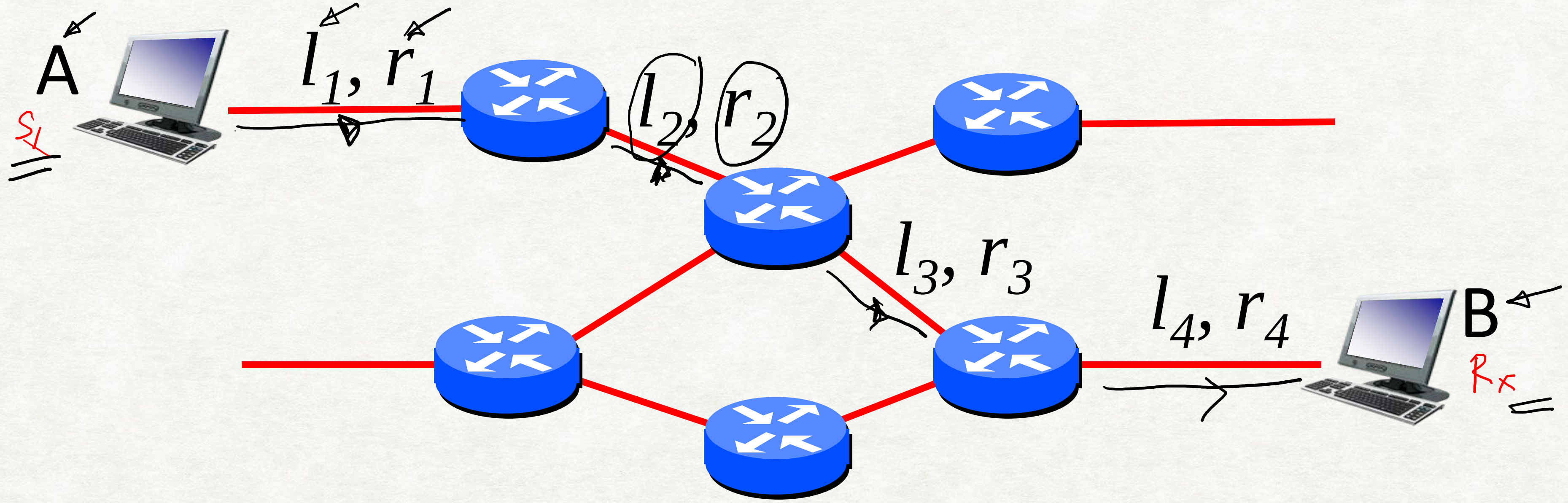
The time from when the first bit is transmitted until the last bit arrives.



transmission

$$\underline{t} = \underline{t_p} + \underline{t_l} = \frac{p}{r} + \frac{l}{c}$$

End-to-end delay



$$\text{End-to-end delay, } t = \sum_{i}\left(\frac{p}{r_i} + \frac{l_i}{c}\right)$$

$i = 1, 2, 3, 4$

Exercise Time!

Suppose that a packet has 1500 bytes. It needs to be transmitted, over a fiber link that has a bandwidth of 12 Gbps.

(a) What is the transmission delay (the time needed to transmit all of a packet's bits into the link)?

$$t_p = \frac{1500 \times 8}{12 \times 10^9} = \frac{12000}{12 \times 10^9} = 10^{-6} = 1 \mu s$$

Exercise Time!

Suppose that a packet has 1500 bytes. It needs to be transmitted, over a fiber link that has a bandwidth of 12 Gbps.

(a) What is the transmission delay (the time needed to transmit all of a packet's bits into the link)?

$\underline{1\,\mu s}$

(b) What is the maximum number of packets per second that can be transmitted by the link?

$1\,\mu s \rightarrow 1$ packet

1 sec $\rightarrow 10^6$ packets $=$ 1 million packets

Exercise Time!

Suppose that a packet has 1500 bytes. It needs to be transmitted, over a fiber link that has a bandwidth of 12 Gbps.

(a) What is the transmission delay (the time needed to transmit all of a packet's bits into the link)? $\rightarrow t_p = 1\,\mu s$

(b) What is the maximum number of packets per second that can be transmitted by the link? $1s \rightarrow 10^6$ packets

(c) How long will a bit take to travel 1,000km in the link?

$$t_l = \frac{1000 \times 10^3\, m}{2 \times 10^8\, m/s} = 0.5 \times 10^{-2}\, s = 0.005\, s = 5\, ms$$

Propagation Delay

(d) What is the total time to send a packet over this link?

$$t_{total} = t_p + t_l = 1\,\mu s + 5\, ms = 5.001\, ms$$

Exercise Time!

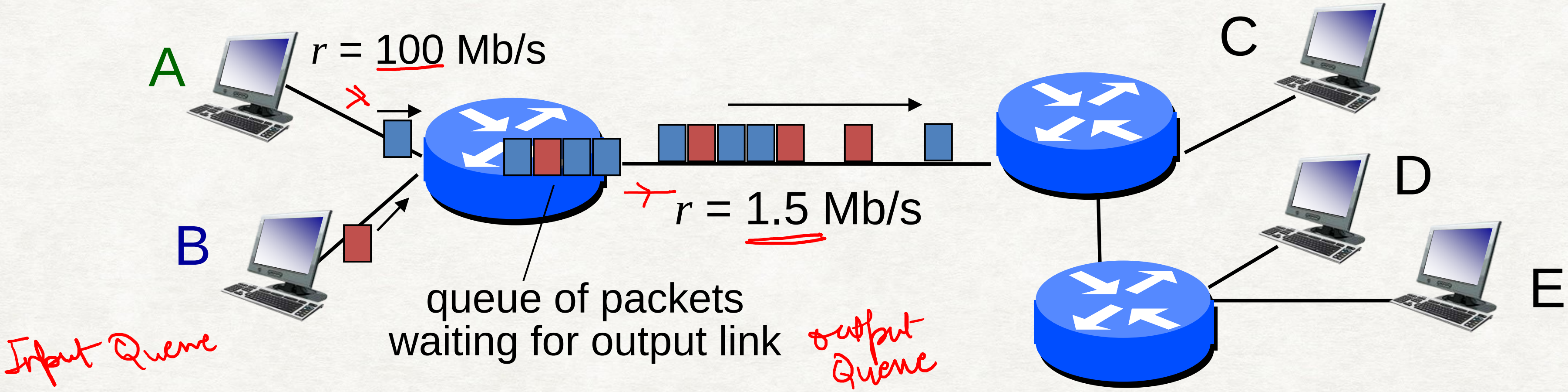
Suppose that a packet has 1500 bytes. It needs to be transmitted, over a fiber link that has a bandwidth of 12 Gbps.

(a) What is the transmission delay (the time needed to transmit all of a packet's bits into the link)?

(b) What is the maximum number of packets per second that can be transmitted by the link?

(c) How long will a bit take to travel 1,000km in the link?

Packet Switching: queueing delay



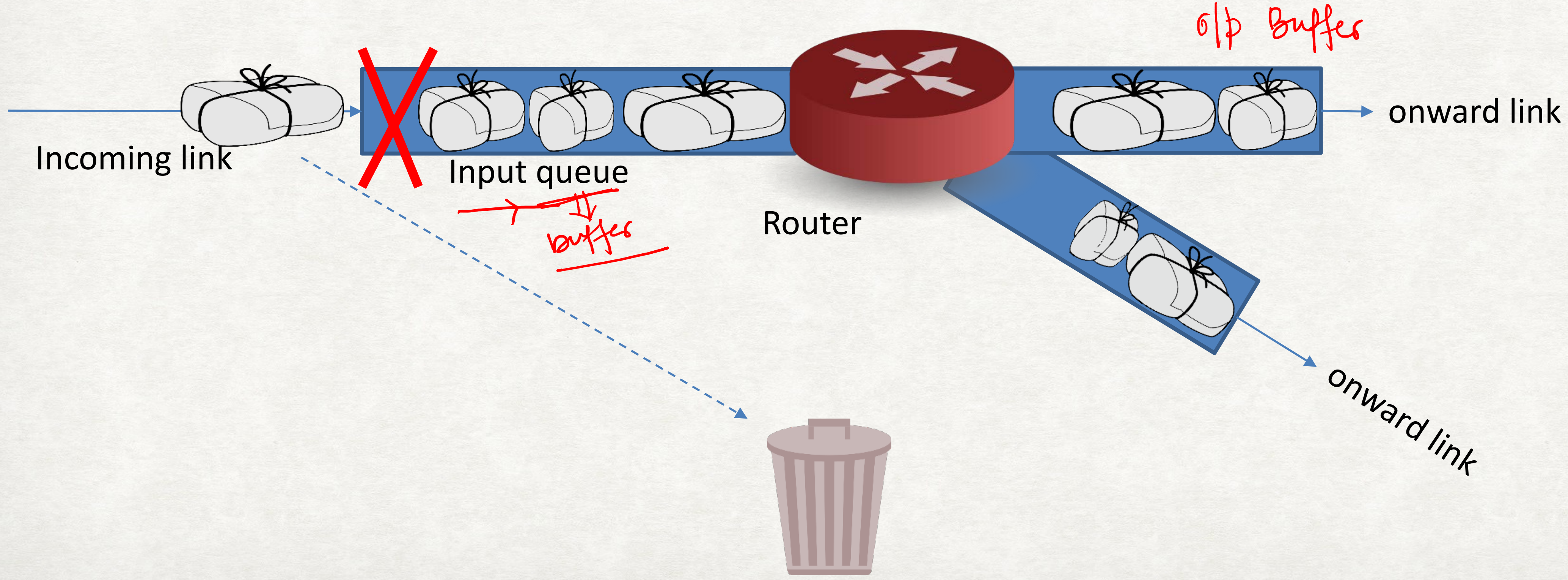
Queuing and loss:

- If arrival rate (in bits) to link exceeds transmission rate of link for a period of time:
  - packets will queue, waiting to be transmitted on link
  - packets can be dropped (lost) if router memory (buffer) fills up

Reference : *Computer Networking: A Top Down Approach by Kurose & Ross*

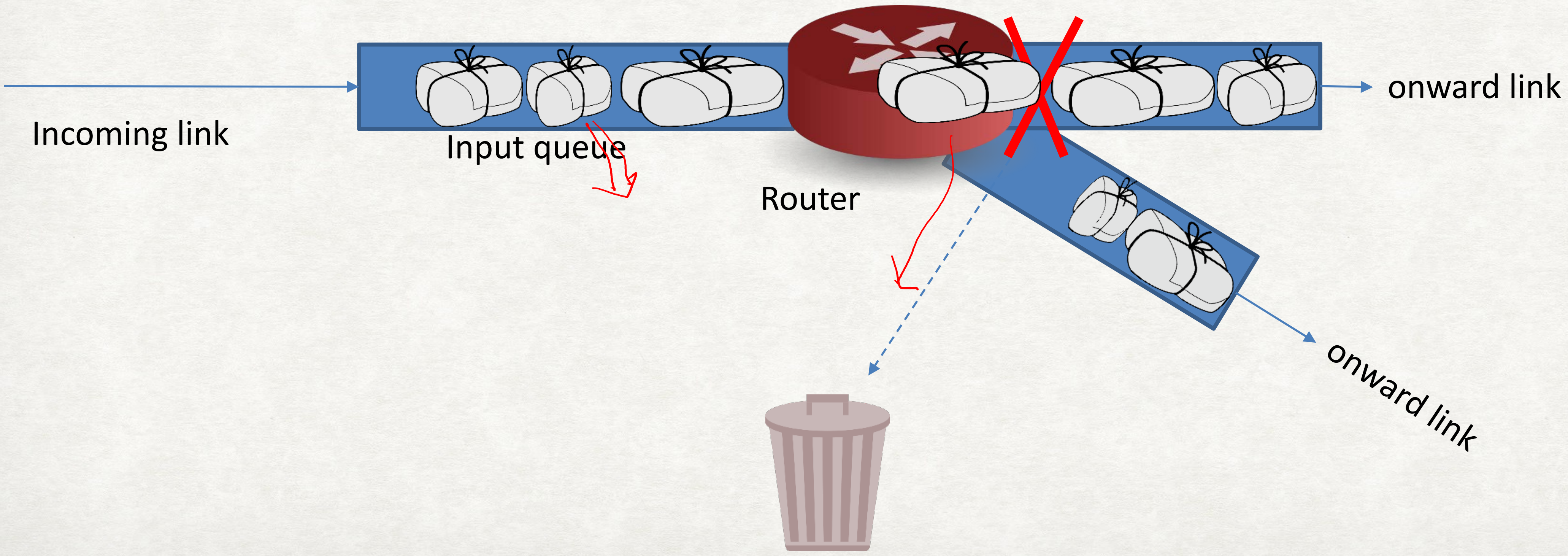
Packet Switching: Packet Loss

- Packets that arrive on a queue that doesn't have enough space left to store them will be *dropped.*
- Dropping a packet means that the router deletes the packet from its memory. It is then permanently lost.

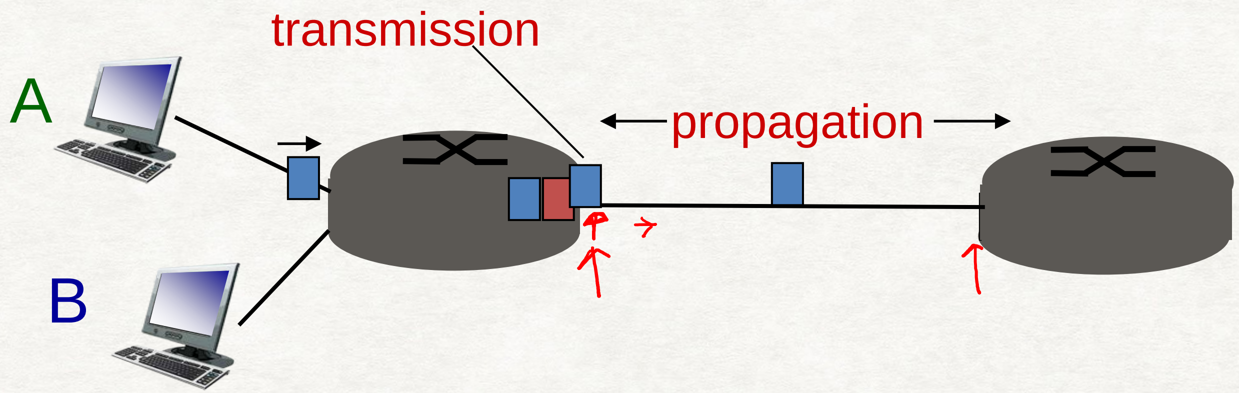


Packet Switching: Packet Loss

- Queue drops can happen on both input and output queues.
- The probability that a packet can still fit in to an almost full queue decreases as the packet size grows.
- As a result, longer packets are more likely to be dropped.



Four sources of packet delay



d_{trans}: transmission delay:

- p: packet length (bits)
- r : link *bandwidth* (*bps*)
- $d_{trans} = p/r$

d_{prop}: propagation delay:

- l: length of physical link
- c: propagation speed in medium (~2×10^8 m/sec)
- $d_{prop} = \dfrac{l}{c}$

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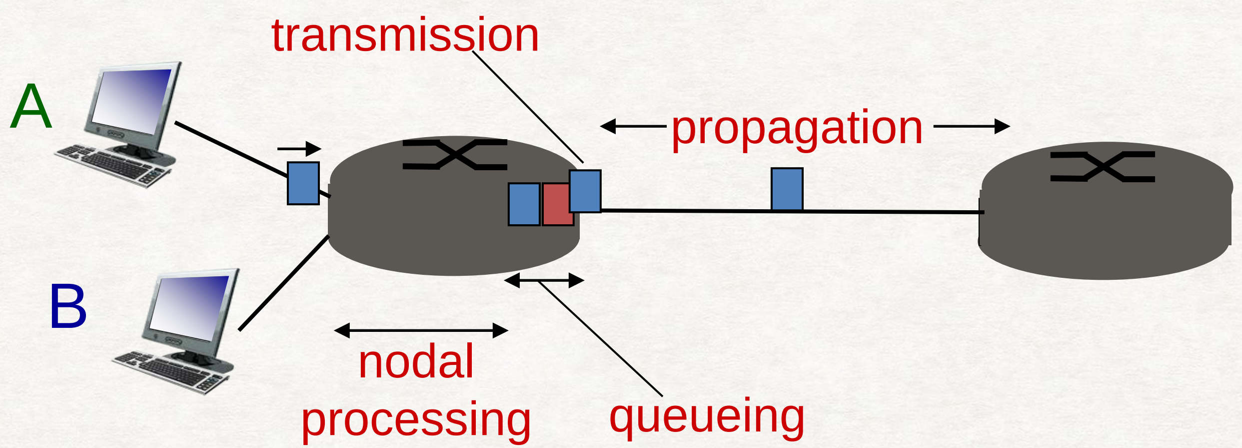
Four sources of packet delay



$$d_{nodal} = d_{proc} + d_{queue} + d_{trans} + d_{prop}$$

"d_{proc}: nodal processing"

- check bit errors
- determine output link
- typically < msec

"d_{queue}: queueing delay"

- time waiting at output link for transmission
- depends on congestion level of router

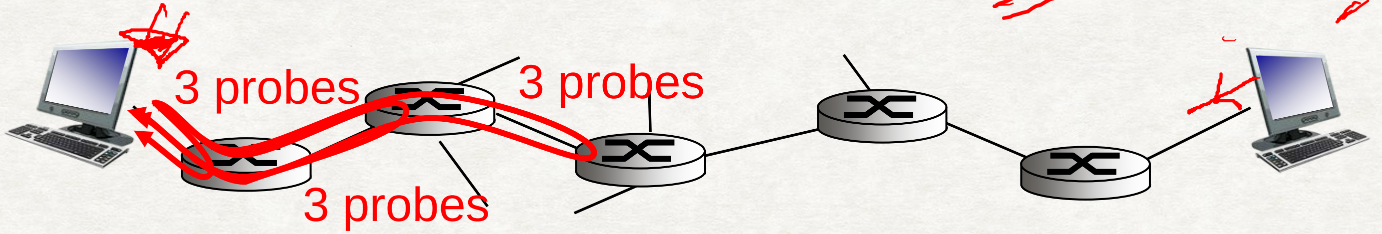
Reference : *Computer Networking: A Top Down Approach by Kurose & Ross*

"Real" Internet delays and routes

What do "real" Internet delay & loss look like?

`traceroute` program: provides delay measurement from source to router along end-end Internet path towards destination.

3 pkts TTL = 1
3 pkts TTL = 2
3



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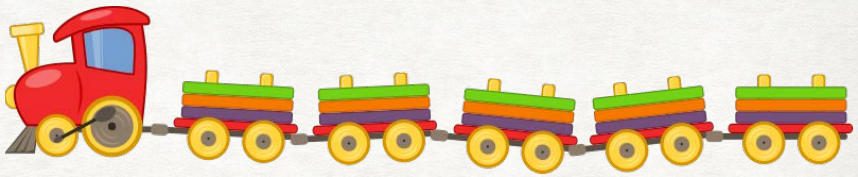
Damage: Bit errors in packets

0 1

1 0



Think of a packet as a train, with the bits being the locomotive and the carriages – one locomotive / carriage = one bit

What's the probability that a train from Auckland to Wellington will arrive if each carriage has a certain probability P of breaking down?

- What is the probability that a packet we send will arrive, if for each bit, there is a probability that the receiver will receive it incorrectly?

Answer: The more carriages the train has, the higher the probability that it will break down along the way

- The longer our packet is, the higher the probability that there will be bit errors

The better the track and the stronger the carriages, the lower the probability that a carriage will break down – can run longer trains

- The stronger our transmitted signal is, and the less the interference in the channel, the fewer bit errors we will get.

Right-sizing packets

Large packets create unfairness on the network.

However, if the packets are too small, they're not efficient.

The "best" packet size depends on a lot of factors:

- The higher the network bit rate, the easier it is to achieve fairness even with larger packets.
- The lower the bit error probability of a network, the more likely that a packet will not be affected by bit errors.
- The smaller a packet, the lower the likelihood that it will be dropped from the end of a near-full queue.

Summary

Data in the Internet travels in packets.

Packets consist of a header and a payload.

- A header is said to encapsulate the payload.

Packets suffer from queuing delays and loss in the Internet.

Packets also can be damaged due to bit errors during transmission.

Very large or very small maximum packet sizes are not a good choice.

- Very large packets are unfair and carry a higher risk of loss or damage.
- Very small packets are inefficient.